Voice Products

5

VOICE

Carriers and enterprises are looking for ways to reduce costs by transporting their voice traffic with greater bandwidth efficiency. This can be achieved by using state-of-the-art voice trunking and compression algorithms, which maintain toll-quality voice, and by converging voice and data traffic over packet switched networks, or by using voice over IP standard solutions.

Voice trunking

Long distance and international leased lines are still quite expensive in many parts of the world, especially satellite-based communications. Reducing the amount of bandwidth required for voice transmission can significantly reduce OpEx. RAD's voice trunking gateways use advanced voice compression algorithms, as well as silence suppression and RAD's unique TDMoIP multiplexing, to reduce the amount of bandwidth required for voice transmission by up to 16 to 1. This translates into significant reduction in OpEx, while maintaining toll-quality voice, without compromising signaling, fax and telephony features.

Benefits of RAD's voice trunking solutions

- Significant bandwidth reduction (up to 16:1) translates into significant cost savings
- High quality voice
- Full transparency to signaling and telephony features eliminates the need for additional capital expenditure or retraining of staff
- Future-proof solution, equipped with both TDM and IP network links
- Reduce initial investment with scalable products that are simple to install and maintain

Typical applications

Inter-MSC trunking – Typically hundreds of long haul links are used to transport traffic between MSCs (E-channel) in meshed, star or mixed network topologies. Limiting the number of links translates into immediate cost savings. RAD's Vmux/Gmux voice trunking gateways use advanced voice compression technologies to significantly reduce the number of required leased lines. This savings translates into fast payback.

Offshore call centers – The cost of international leased lines is still a large contributor to the overall OpEx involved in operating an offshore call center, especially one with thousands of seats. Using RAD's Vmux voice trunking gateways, providers of offshore call center services can significantly reduce their operational costs, without degrading the level of service they provide.

PBX extension over satellite – Providing voice services to remote/mobile platforms usually involves satellite communications, which significantly increases OpEx for the oil/gas, maritime, broadcasting, emergency response, and government/military applicaitons. RAD's Vmux voice trunking gateways increase the capacity and reduce the cost of extending voice services over satellite.

VoIP

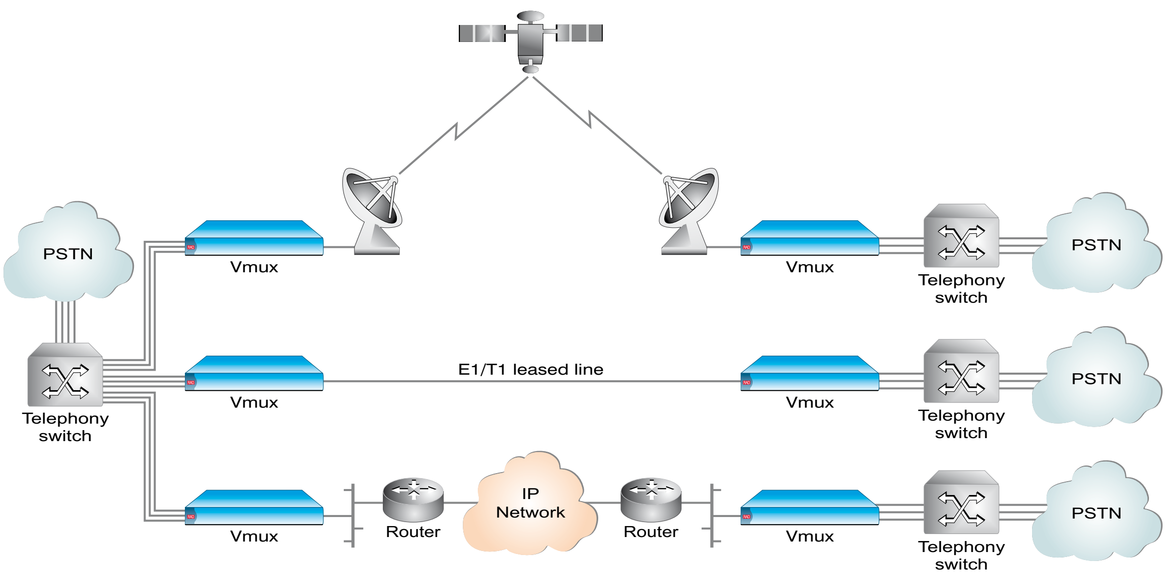
The rapid advancement in broadband deployment is creating new opportunities for both service providers and enterprises to benefit from converged voice and data networks and cost-cutting Voice over Internet Protocol (VoIP) services. The RAD VoIP System (RVS) enables seamless transition to IP telephony, leveraging existing equipment to tap into the savings and advanced technology available from VoIP communications. An invest-as-you-grow platform, the RVS has been designed to enable VoIP services at a low initial entry cost, minimizing total cost of ownership. Consisting of various customer premises equipment (CPEs), a high availability Class 5 softswitch and powerful management system – as well as complementing application servers, such as voice mail, IVR, billing and more – RVS is an out-of-the-box solution that enables providers to launch a VoIP service in days. A SIP-based, carrier-grade system, the RVS has been engineered for quality, reliability and scalability – key factors when rolling out a VoIP service.

Benefits of RAD's VoIP System

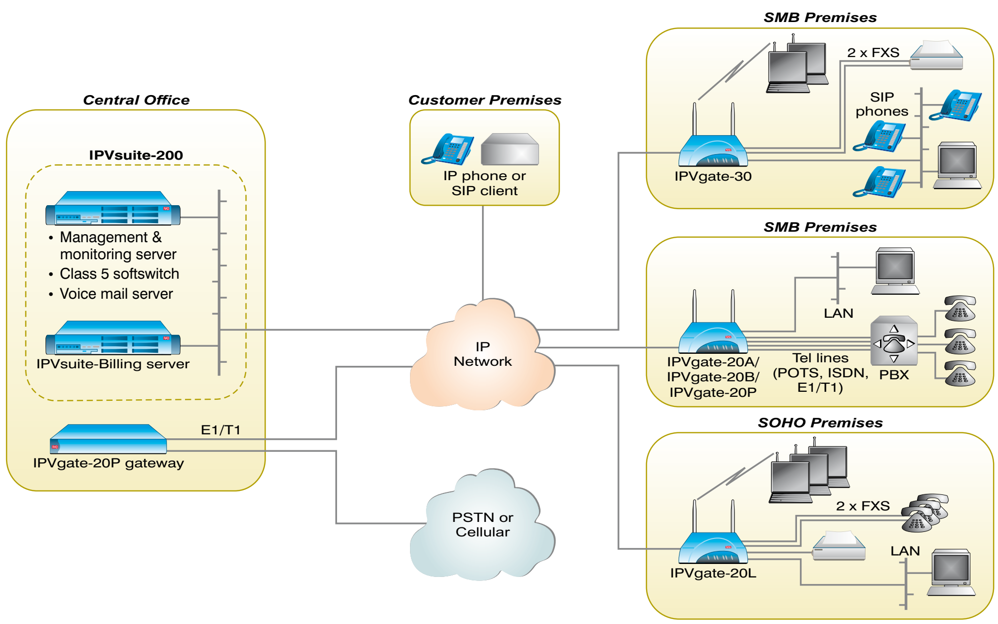
- Fully integrated package controlled by a single management system
- Zero time to service
- Low initial investment – "pay-as-you-grow" model
- Unique remote analysis tools save on truck rolls
- Compatible with all standard SIP clients

Typical application

VoIP telephony access – Traditional voice access services were provided to business and residential customers by TDM Class 5 switches. The cost of TDM resources are extremely high compared to an IP-based solution. RVS provides all the essential parts of a VoIP access service: management and provisioning, softswitch and a variety of smart CPEs that enable the telephony service provider and the end user smooth migration to IP-based access services (voice and data) with tremendously low costs and high availability of service.



Voice trunking over TDM/IP links



Business-grade telephony access for alternative carriers and ISPs

Vmux-2100

Voice Trunking Gateway

- **Compresses up to 16 full E1/T1 voice lines over a single E1/T1, serial or IP link**
- **Up to 16:1 voice compression, along with silence suppression and TDMoIP multiplexing for maximal bandwidth savings**
- **High quality voice**
- **More bandwidth efficient than standard VoIP**
- **Transparent to all signaling protocols and telephony features**
- **Enhanced relay mechanisms for fax/ modem/DTMF/special tones**
- **Compact, scalable and simple to deploy**
- **Local and remote management via ASCII terminal, Telnet or RADview-SC/ Vmux**

For latest updates visit www.rad.com

RAD's Vmux-2100 voice trunking gateway employs powerful voice compression algorithms, reducing leased line costs and increasing the efficiency of the IP network. Vmux-2100 compresses up to 16 full E1/T1 lines (496/384 voice channels) over a single E1/T1, serial or IP uplink, enabling enterprises, mobile operators and service providers to save costs by leasing fewer lines to transport their voice payloads.

The Vmux-2100 is especially suited for satellite connectivity, remote call centers, 2G and 3G cellular backhaul, international voice trunking, wireless Local Loop, and rural telephony. Vmux may be used in narrowband applications, wherever there is a need to minimize bandwidth for voice transmissions, over any media (for example, TDM or IP satellite links).

Voice compression reduces line costs

Vmux-2100 uses G.723.1, G.729A and G.711 voice compression algorithms for optimal cost/ performance. It maintains toll-quality voice while achieving the highest compression ratio for voice transmission over TDM and IP networks. Voice activity detection and silence suppression allow Vmux-2100 to dynamically utilize bandwidth for voice traffic and fax or modem relay, resulting in very efficient bandwidth usage over fewer lines, while signaling information is transmitted separately.

Uses less bandwidth than VoIP

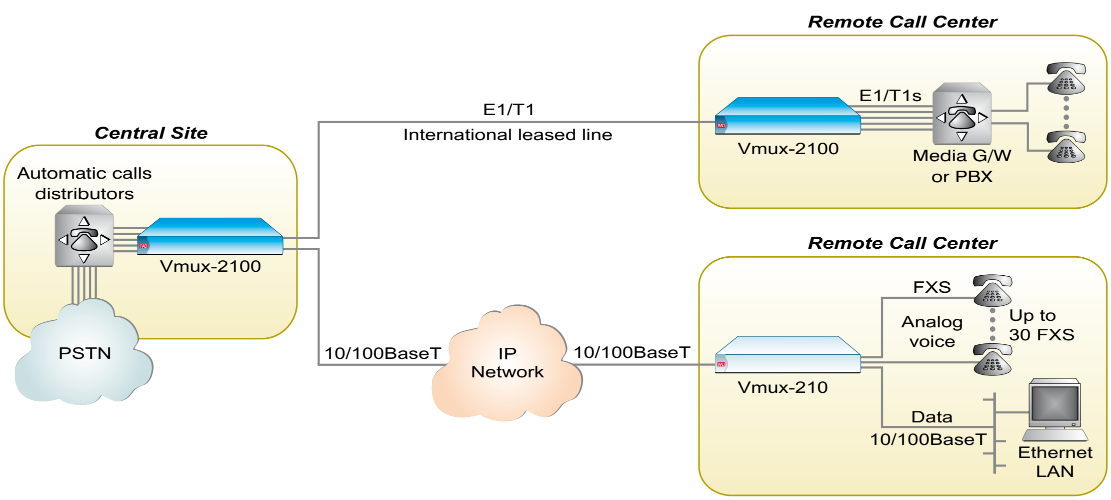
TDMoIP capabilities improve bandwidth utilization and provide a real migration path to IP. Separate TDM and Ethernet uplinks enable simultaneous transmission over both TDM and IP networks. With lower overhead than VoIP systems, Vmux frees 60 percent more bandwidth for additional voice or data, which is crucial on costly or limited bandwidth links.

Converged services save equipment and maintenance costs

Network integration is the key to maximum savings in the wide area network. Vmux-2100 converges voice, fax and Ethernet traffic over the same network link. By combining voice/fax capabilities with Ethernet data traffic over a single delivery network, operators can save significantly on the costs of running their networks.

Space-saving unit

The compact, 1U-high Vmux-2100 is a modular unit that can be installed in 19-inch racks. It has the smallest footprint of any TDM voice compression or VoIP product of equivalent capacity. The unit can be configured with up to four voice compression modules, up to four channelized TDM modules and up to two power supply modules. All modules are plug-in and field-replaceable. The power supply and voice compression modules are hot-swappable.



Offshore call centers

Vmux-210

Analog Voice Trunking Gateway (Compressed Channel Bank)

Vmux-210 is a remote voice trunking gateway for both IP and leased line TDM networks, providing LAN and compressed voice services for corporate applications that require a large number of analog lines for POTS or fax connection. It is a customer-located device that complements RAD's larger modular Vmux-2100 system equipped with E1/T1 voice interfaces.

Voice compression for analog lines

Vmux-210 compresses voice traffic and transports it over a serial link, E1/T1 link, or a 10/100BaseT IP uplink. The device employs G.723.1, G.729 Annex A and G.711 compression algorithms together with RAD's unique TDMoIP multiplexing, including transparent CAS.

Voice activity detection (VAD) and silence suppression

Voice activity detection (VAD) and silence suppression allow Vmux units to dynamically allocate bandwidth for voice traffic. Efficient bandwidth usage leaves more bandwidth for data transport. LAN data traffic can be controlled with rate limiting.

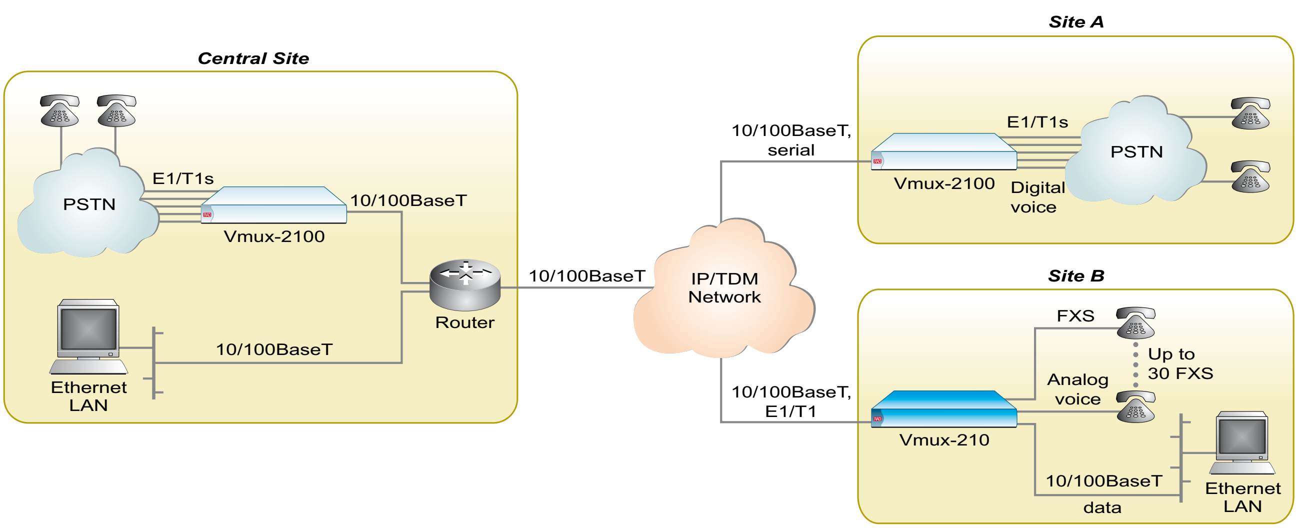
The gateway is transparent to all signaling protocols and telephony features.

Vmux-210 can be configured and monitored via a local ASCII terminal, Telnet or RADview-SC/Vmux.

Vmux-210 is a compact, 1U-high, 19-inch wide unit that can be mounted in standard 19-inch racks. The voice interface options include 12, 15, 24, or 30 FXS analog ports. The unit is available with either AC or DC power supply.

- **Compresses up to 30 FXS voice lines over an E1/T1, serial or IP link**
- **Uses voice compression, silence suppression and TDMoIP multiplexing for maximal bandwidth savings**
- **High quality voice**
- **More bandwidth efficient than standard VoIP**
- **Compatible with all types of VSATs**
- **Transparent to all signaling protocols and telephony features**
- **Additional user LAN port with voice/data prioritization**

For latest updates visit www.rad.com



Data and compressed digital/analog voice over IP network

Vmux-110

Remote Voice Trunking Gateway

- Compresses four or eight FXS/FXO/E&M voice lines or a single E1/T1 voice line over an E1/T1, serial or IP link
- Up to 16:1 voice compression, combined with silence suppression and TDMoIP multiplexing for maximal bandwidth savings
- High quality voice
- More bandwidth efficient than standard VoIP
- Compatible with all VSATs
- Transparent to all signaling protocols and telephony features
- Additional user LAN port with voice/data prioritization

For latest updates visit www.rad.com

The Vmux-110 is a customer-located device that complements the Vmux system, providing LAN and compressed voice transmission over both TDM and Ethernet-based networks. Employing powerful voice compression algorithms as well as TDMoIP technology, the Vmux-110 can compress a full E1/T1 or four or eight analog lines, leaving more bandwidth for data transport.

Reduces line costs

It supports four or eight FXS/FXO/E&M ports or a single full or fractional E1/T1 voice port. Vmux-110 compresses voice traffic and transports it over an n x 64 kbps, E1/T1 or IP link. The device employs G.723.1, G.729 Annex A and G.711 compression algorithms together with RAD's unique TDMoIP multiplexing, and is transparent to all signaling protocols and LAN.

Silence suppression improves bandwidth utilization

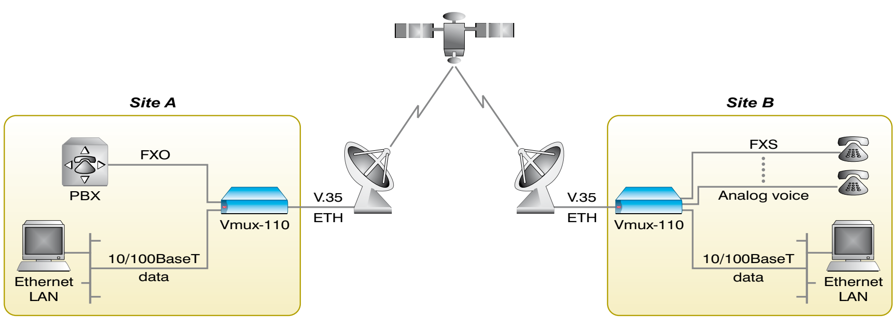
Voice activity detection and silence suppression allow these Vmux units to dynamically allocate bandwidth for voice traffic. This results in very efficient bandwidth usage, leaving more bandwidth for data transport, further controlled with rate limiting capabilities.

Ethernet port for data

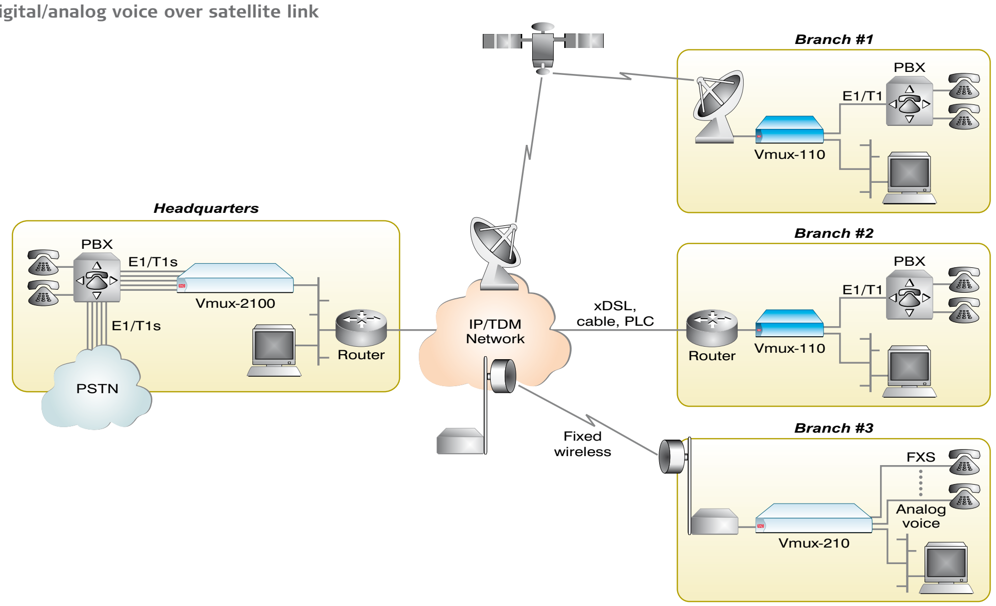
For data connectivity, the customer-located Vmux-110 features a second 10/100BaseT port and an Ethernet switch for integrating the user LAN traffic with the compressed voice over an IP, serial or E1/T1 uplink.

Management

Local and remote management for the Vmux-110 can be performed via ASCII terminal, Telnet or RADview-SC/Vmux. The Vmux-110 is a compact, 1U-high, half 19-inch wide unit that can be mounted in a 19-inch rack. The unit is available with either an AC or DC power supply.



Data and compressed digital/analog voice over satellite link



PBX and LAN extension over limited bandwidth infrastructure

Gmux-2000

Carrier Voice Trunking Gateway

The Gmux-2000 is a carrier-class, modular voice trunking gateway that provides a cost-effective high capacity solution for reducing the bandwidth required for voice transmission over TDM, IP or MPLS networks.

Using state-of-the-art voice compression algorithms, as well as voice activity detection, silence suppression and RAD's unique TDMoIP multiplexing, Gmux-2000 can reach a maximum compression ratio of 16:1, transmitting up to 112 E1/T1 links over as few as seven E1/T1s, or over a single Gigabit Ethernet link. Alternatively, Gmux-2000 can compress voice coming directly from an STM-1/OC-3 voice trunk.

By optimizing signaling channels (SS7, PRI, etc.) Gmux-2000 further reduces overall bandwidth.

Gmux-2000 maintains a high quality of voice, while ensuring continued support of inband telephony features, such as fax, modem, IVR, and others.

When the Gmux-2000 is deployed opposite the Vmux-2100, Vmux-110 or Vmux-210, it offers a complete, cost-effective carrier-class voice trunking solution.

Modules

Gmux-2000 is a 6U-high chassis, mountable in a 19-inch ETSI or ANSI rack, and housing the following modules:

- Up to two PSN network uplink modules or inband management modules
- Two control modules
- Three AC or DC power supply modules
- Seven I/O modules (voice compression modules or STM-1/OC-3 interface modules)
- A cooling fans module

Gigabit Ethernet network modules, each with a pair of redundant Gigabit Ethernet (GbE) ports, support Ethernet IEEE 802.3ad, 802.1Q (VLAN tagging) and 802.1p (priority bits).

Voice compression modules perform compression and processing of E1/T1 traffic flows, and transmit the compressed voice over the E1/T1 main link ports on the module itself, or through the internal bus to the GbE network module. Each voice compression module is capable of handling 12 or 16 incoming E1/T1s (ordering option). These modules function as server modules, meaning they can receive the voice directly from the PBX/MSC (via a direct Telco cable), or they can process voice coming in from an STM-1 module. The voice compression modules are capable of functioning in point-to-multipoint topologies as well.

STM-1/OC-3 interface modules perform SDH/SONET multiplexing/demultiplexing of channelized STM-1/OC-3 trunks into separate, internal E1/T1 circuits (for processing by the voice compression modules). Each STM-1/OC-3 interface module provides a pair of ITU-T G.703 coax or G.957, G.958 fiber optic links, supporting 1+1 redundancy according to ITU-T G.783.

Control modules provide the SNMP and management interfaces for configuration and control of the entire Gmux-2000 system. Each control module provides redundant interfaces for connecting external G.812 station clocks and for alarm inputs/outputs.

Power supplies are hot-swappable plug-in modules. Up to three AC or DC power supply modules can be installed for load sharing and redundancy. Two modules are required to drive a fully equipped Gmux-2000 system.

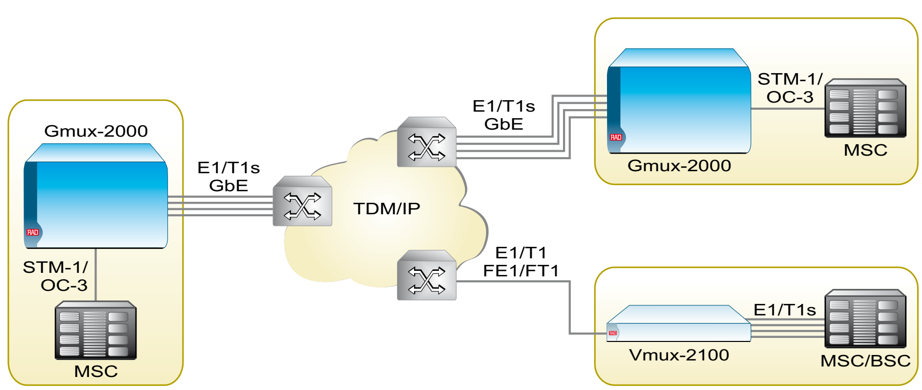
Service center management and security

The Gmux-2000 features flexible management capabilities, including local management via an ASCII terminal (RS-232). In addition, remote management can be performed either inband or out-of-band, using one of the network ports or the dedicated management port, while maintaining separation between management and user traffic via the use of VLANs. Advanced FCAPS (Fault, Configuration, Accounting, Performance, Security), service provisioning and diagnostic tools are provided by RADview-SC/Vmux, RAD's network management system, via an SNMP-based GUI.

The Gmux-2000 also supports a variety of configuration access channels, including CLI over Telnet, SNMP, and TFTP. Incorporated security features include Secure Shell (SSH), Secure FTP (SFTP), SNMPv3, and RADIUS, as well as management access control list (ACL).

- **Supports Inter-MSC 24&34 networks**
- **Compresses up to 112 full E1/T1 voice lines or a single STM-1/OC-3 voice line over E1/T1, SDH/SONET or GbE links**
- **Up to 16:1 voice compression, combined with silence suppression and TDMoIP multiplexing for maximal bandwidth savings**
- **Modular chassis with full hardware redundancy**
- **High quality voice**
- **Transparent to all signaling protocols and telephony features**
- **Additional optimization of signaling channels**
- **Local and remote management via ASCII terminal, Telnet or RADview-SC/Vmux**

For latest updates visit www.rad.com



Voice trunking over multiple E1/T1 streams in a cellular network

RAD VoIP System (RVS)

VoIP Solution for Service Providers

- **Fully integrated system**
- **Central management tool ensures low total cost of ownership (TCO)**
- **Record rollout time**
- **Highly scalable architecture produces quick ROI**
- **Unique remote analysis tools save on truck rolls and ensure low OpEx**
- **SIP-based, compatible with all standard SIP clients**
- **Multilayered quality of service (QoS)**
- **Delivers toll-quality voice calls**
- **Type of Service (ToS) tagging to assign priority to VoIP packet traffic**

For latest updates visit www.rad.com

The RAD VoIP System (RVS) is a carrier-grade, SIP VoIP solution for service providers to roll out next-generation VoIP services to business and residential customers with broadband infrastructure in minimal time with a low initial investment.

The RVS's "pay-as-you-grow" architecture delivers quick ROI, allowing providers of all sizes to launch service with a small number of VoIP subscribers and invest more with the expansion of the customer base.

The RVS is an opportunity for service providers to tap into new revenue sources, offering business and SOHO customers easy migration to cost-cutting VoIP telephony without replacing their existing telephony equipment.

Integrated solution for high quality VoIP services

Engineered to provide superior voice quality and call integrity, the fully integrated RVS is comprised of the IPVsuite-200 Class 5 SIP softswitch and provisioning system, IPVgate customer premises VoIP gateway routers, an IP-PBX for enterprises, and the IPVsuite-Billing prepaid and postpaid billing server.

The RVS is an inclusive system connecting the central office softswitch to customer premises analog telephones and fax machines, and analog, ISDN, and IP PBXs. Enterprises benefit from LAN connectivity via a 10/100BaseT Ethernet port. For data communications between headquarters and remote offices, a built-in VPN server/client creates transparent connectivity of enterprise LANs into a single virtual LAN.

Integral to the system are an ADSL modem and router, firewall, IPSec VPN, and encrypted phone/fax P2P VoIP calls.

Optional Wi-Fi access point capability and a full-featured IP-PBX complement the offering for the more demanding SMB environment.

SIP compliant

The RVS is a SIP-based system fully compatible with leading SIP end units, including gateways, IP phones and IP-PBXs. The system enables connection to any other standard SIP client after its authentication and registration.

Central management reduces OpEx

System repairs and upgrades are handled remotely from central sites, saving money for providers and keeping total cost of ownership to a minimum. The RVS's multilayered quality of service (QoS) includes call prioritization and bandwidth optimization, enabling the delivery of high quality calls even over low-cost ADSL connections.

The RAD VoIP System:

IPVgate-20A: SMB analog VoIP gateway router

IPVgate-20B: SMB BRI VoIP gateway router

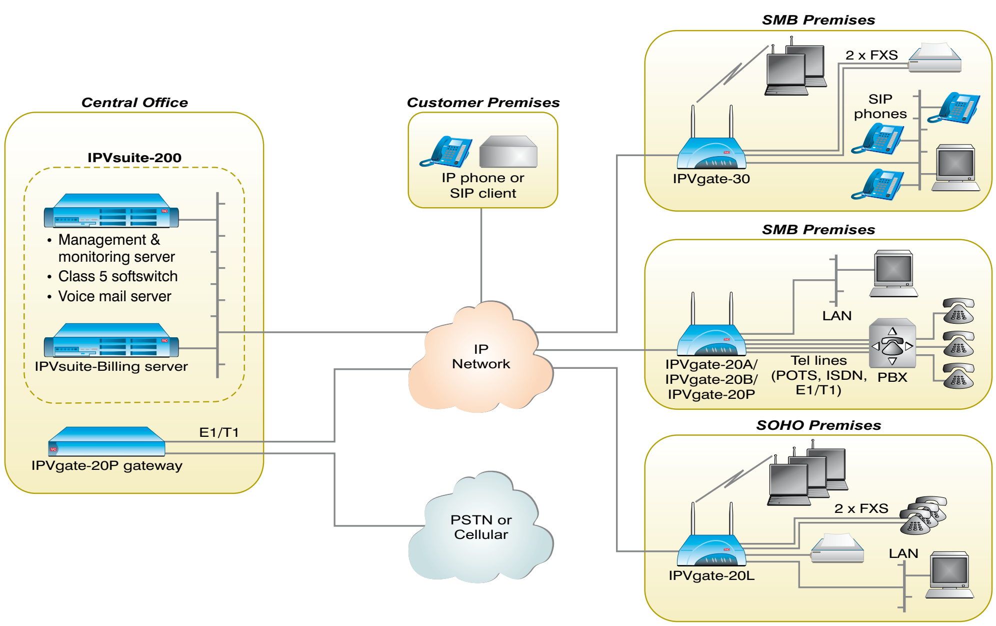
IPVgate-20L: SOHO analog VoIP gateway router

IPVgate-20P: PRI VoIP gateway router (can also function as media gateway)

IPVgate-30: SMB IP-PBX and VoIP gateway router

IPVsuite-200: SIP Class 5 softswitch and provisioning system

IPVsuite-Billing: Prepaid and postpaid billing server



IPVsuite-200

Class 5 SIP Softswitch and Provisioning System

The IPVsuite-200 Class 5 SIP softswitch and provisioning system is a carrier-class, high-performance Session Initiation Protocol (SIP) solution for integrating and deploying next-generation VoIP, data and multimedia services over a packet switched network.

Built-in management and provisioning

The IPVsuite-200 includes a built-in provisioning and management system, providing a comprehensive central office VoIP service solution for service providers. The system enables providers to view, manage, monitor, and configure various CPEs and applications installed in the network, thereby reducing the integration and operational costs of the CPEs and drastically minimizing truck rolls.

Easy scalability for expansion of VoIP services

The highly scalable SIP softswitch, with a capacity starting at 400,000 busy hour call attempts (BHCA), and the built-in management system enable providers to roll out high quality, carrier-grade VoIP services in record time and expand easily with the addition of subscribers.

Advanced services

The IPVsuite-200 features enhanced call routing as well as fallback management capabilities. The advanced Class 5 telephony services offered by the IPVsuite-200 SIP softswitch include enhanced voice mail, auto-attendant, call forwarding, DND (do not disturb), ACR (anonymous call rejection), incoming/outgoing call screening, emergency numbers (by ZIP code), and toll-free numbers.

The system is compatible with leading VoIP media gateways and SIP end units, including soft phones, IP phones and IP-PBXs. IPVsuite-200 enables the connection of any other standard SIP client after its authentication and registration. The SIP client will also benefit from Class 5 services provided by the switch.

Integrated billing

The IPVsuite-200 softswitch generates Call Data Records (CDR), featuring a fully integrated SIP application server for prepaid services. RAD's billing solution, IPVsuite-Billing, handles all prepaid and postpaid accounting.

- **Low total cost of ownership (TCO)**
- **Quick and easy installation for record speed service rollout**
- **End-user Web page for subscriber self-provisioning**
- **Robust cluster architecture for carrier-class high availability**
- **High scalability starting from 400,000 BHCA switching performance**
- **Powerful real-time monitoring and troubleshooting for reduced truck rolls**
- **High quality auditing, error and call-quality reports**
- **Compatible with standard SIP clients such as IP phones and IP-PBXs**

For latest updates visit www.rad.com

Monitoring system

IPVsuite-Billing

Prepaid & Postpaid Billing Server for SIP VoIP Networks

- **Quick and easy installation for fast service rollout**
- **Highly scalable for easy expansion of customer base**
- **Handles prepaid and postpaid accounting for VoIP services**
- **Fully integrated SIP application server for prepaid services**
- **Supports phone-to-phone, PC-to-phone and PC-to-PC services**
- **Supports calling cards and PIN code management**
- **User-friendly Web interface for easy service provisioning**
- **Robust architecture for carrier-class availability**

For latest updates visit www.rad.com

With the evolution of Internet telephony, incumbent, alternative carriers and ISPs are all offering low cost, high quality VoIP services to enterprise and residential customers. A powerful billing system is essential to capitalize on growing VoIP revenues.

IPVsuite-Billing, part of the RAD VoIP System (RVS), provides a comprehensive solution for postpaid and prepaid billing of SIP VoIP services. It is perfectly suited for fast deployment of advanced phone-to-phone, PC-to-phone and PC-to-PC services. The solution enables the provider to rapidly deploy competitive new services and rating plans.

Pay-as-you-grow

IPVsuite-Billing is a highly scalable system, designed according to the "pay-as-you-grow" model. This enables the provider to launch a VoIP operation with a reasonable budget, ensuring that operational costs are proportional to revenues, while enabling expansion of licenses and hardware with growth.

Advanced account management

The real-time interaction of the SIP application server with the network elements enables IPVsuite-Billing to control the call and cut it off as the customer's balance bottoms out, preventing leakage and securing the ROI.

For postpaid customers, comprehensive invoicing, account receivables and collection capabilities are part of the solution, supporting residential, enterprise and SMB business models.

IVR options

Fully integrated with RAD's IPVsuite-200 Class 5 SIP softswitch and provisioning system, the IPVsuite-Billing prepaid server includes service applications for zero-stage direct dialing services without IVR, single-stage dialing services with IVR (for registered prepaid subscribers) and dual-stage dialing services with IVR (for prepaid calling cards).

Zero-stage dialing prepaid services include direct destination number dialing, automatic CLI-based authentication, destination and balance-based call authorization, automatic cut-off on balance exhaustion, toll-free calls with optional time limits even when balance is zero, and invocation of SIP error responses which can be translated to special tones or announcement with the use of a dedicated announcement server.

Single-stage dialing prepaid services include special access number dialing, multi-lingual announcements customized per subscriber, per access number or via language selection menu, CLI or PIN-based authentication, optional action menus, destination and balance-based call authorization, toll-free calls with optional time limits even when balance is zero, configurable prompts for balance and time-left announcements, mid-call warning with configurable threshold, long pound disconnect, IVR return on call completion, voucher-based account or calling card recharge, and calling card password change with optional new password playback.

User-friendly Web interface

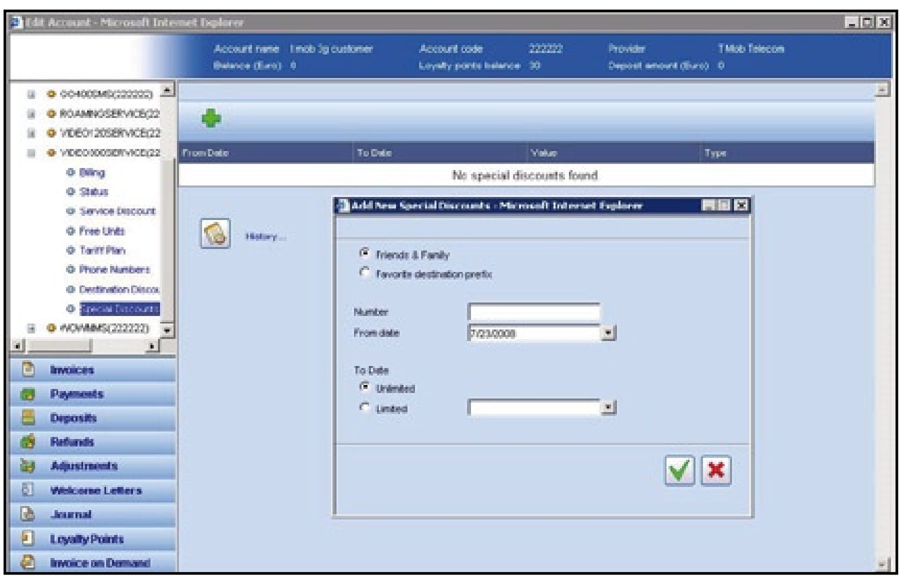
The IPVsuite-Billing's Web interfaces provide up-to-the-minute account information for both the subscribers and customer service representatives (CSRs). Using the WebClient application, CSRs can easily create, search and modify accounts, view account activities, balances and invoices, and manage support tickets.

Customers are able to manage their own accounts, register to new services, and make payments online through Web-based self-care interface. Calling card customers can change their PIN numbers and top-up their balance via the Web or over the phone using an IVR system.

Full billing capabilities, including invoice generation and shipment, account receivables (A/R) management, collection, and general ledger (G/L) interfaces with accounting systems, are inherent to the solution.

Invoicing

IPVsuite-Billing gives providers the possibility to spread the billing and invoicing of the client base throughout the month, ensuring a steady flow of revenue. Multiple billing cycles can be predefined and assigned to customers according to company policy. In addition, the system gives providers the possibility to personalize invoices by adding logos, taglines and special offers within the invoice, with the invoice layout options.



IPVgate-20A, IPVgate-20B, IPVgate-20P

SMB VoIP Gateway Routers

The IPVgate-20A, IPVgate-20B and IPVgate-20P SMB VoIP gateway routers are customer premises equipment (CPEs) for deployment of SIP-based VoIP services to small and medium businesses (SMBs) with ADSL2+ or Ethernet broadband connectivity.

Enterprise communications center

Robust end-routers, interoperable with any standard SIP device, the IPVgate-20A, IPVgate-20B and IPVgate-20P provide full telephony line replacement in addition to broadband access, data communications and enterprise networking. The IPVgate-20P can be deployed both as a VoIP access gateway and a VoIP media gateway.

The SMB VoIP gateway routers connect legacy analog telephones, analog and ISDN PBXs, fax machines, and data networks over broadband to SIP-based IP telephony networks.

The products include built-in ADSL modems, routers, firewalls, and VPN, enabling easy and secure connection between headquarters and branch offices. Fax machines connected to the CPEs also benefit from high quality T.38 support, ensuring reliable fax transmissions.

The built-in VPN IPSec server/client utilizes a 3DES encryption algorithm and up to 10 tunnels per device, creating transparent connectivity of enterprise LANs into a single, virtual one.

Intelligent CPEs deliver high QoS

When operating as part of the RAD VoIP System, the IPVgate-20A, IPVgate-20B and IPVgate-20P work seamlessly with the IPVsuite-200 SIP softswitch's built-in management and provisioning system, providing network statistics and analysis for remote troubleshooting.

Engineered for superior voice quality, the IPVgate products have multilayered quality of service (QoS) mechanisms to ensure integrity of VoIP telephone calls. The units perform Type of Service (ToS) tagging to assign priority to VoIP packet traffic and thereby ensure integrity of VoIP phone calls.

The intelligent IPVgate gateways are equipped with monitoring tools, enabling the central office to remotely configure, provision and manage units in service, thereby keeping OpEx to a minimum.

Interfaces

The IPVgate-20A analog VoIP gateway router connects legacy analog telephones, PBXs, faxes, and data over broadband to SIP-based IP telephony networks. The product connects two or four FXS ports over any broadband Internet connection. Secure wireless networking by Wi-Fi access point (IEEE 802.11b and IEEE 802.11g) is an optional feature.

IPVgate-20B BRI VoIP gateway router connects two or four ISDN BRIs over IP. The IPVgate-20B has an additional BRI (RJ-45) interface, which is connected to the PSTN as a backup.

IPVgate-20P E1/T1 PRI VoIP gateway router connects 30 lines (single PRI port) over any broadband connection. The unit allows up to 30 concurrent VoIP or PSTN calls. The IPVgate-20P has an additional E1/PRI (RJ-45) interface, which is connected to the PSTN as a backup and as an optional clock source.

Availability of service

In case of SIP server failure, the IPVgate VoIP gateways establish peer-to-peer phone calls to ensure high availability of service (AoS).

The IPVgate-20A is available with an integrated ADSL modem, which provides ADSL line monitoring as well as automatic rerouting of all data via a backup router in the event of connectivity failure.

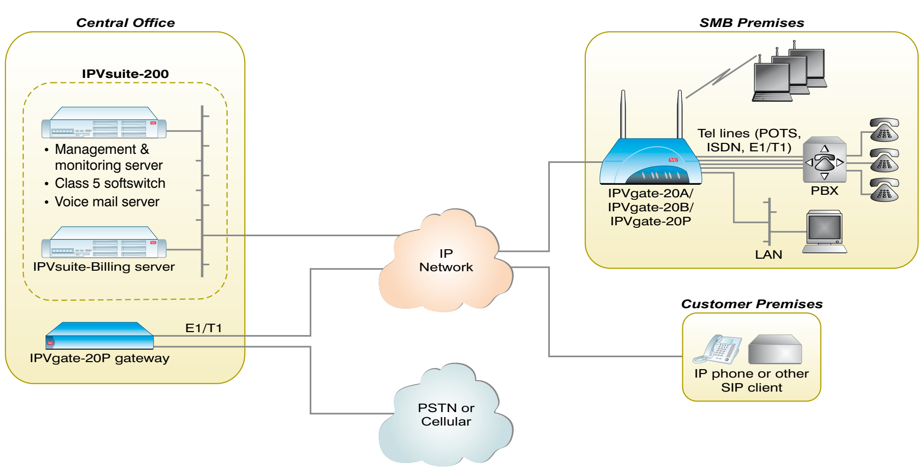
The IPVgate-20B has a BRI backup line and the IPVgate-20P has a PRI backup line to the telecommunications provider that switches all phone calls to the PSTN in the event of a power failure or if the IP network is not functioning or responding.

Advice of charge

The IPVgate-20B and IPVgate-20P are compatible with ISDN PBXs, providing supplementary services, such as advice of charge. The AoC service enables vertical markets, such as hotels and hospitals, to provide billing information to customers while using existing PBXs.

- **Quick and easy installation**
- **Connects 2/4 FXS/BRI/PRI on an ADSL2+ (Annex A/B) or Ethernet connection**
- **Built–in powerful end-router for single-box voice and data communications solution**
- **Built-in VPN (IPSec) server to create enterprise virtual networks**
- **Four Ethernet ports with built-in switch for data connectivity in the LAN (for 20A and 20B only)**
- **WLAN by Wi-Fi access (IEEE 802.11b and IEEE 802.11g) (for 20A and 20B only)**
- **Support for advanced Class 5 call services, including caller ID, call on-hold, call waiting and transfer**
- **Remote management via IPVsuite-200 softswitch ensures low total cost of ownership (TCO)**
- **ADSL line quality monitoring system**
- **Multilayered QoS delivers superior voice quality**
- **Fully interoperable with standards compliant SIP-based equipment**

For latest updates visit www.rad.com



IPVgate-20L

SOHO Analog VoIP Gateway Router

VOIP GATEWAYS

- **Quick and easy installation**
- **Connects two analog FXS ports on an ADSL2+ (Annex A/B) or Ethernet connection**
- **Handles all voice and data connections**
- **Powerful built-in end-router for single-box solution**
- **Four Ethernet ports with built-in switch for data connectivity in the LAN**
- **WLAN by Wi-Fi access (IEEE 802.11b and IEEE 802.11g)**
- **Supports advanced Class 5 call services, including caller ID, call on-hold, call waiting and transfer**
- **Remote management by IPVsuite-200 SIP softswitch ensures low total cost of ownership (TCO)**
- **Multilayered QoS ensures superior voice quality**
- **ADSL line quality monitoring system**
- **Fully interoperable with standards compliant SIP-based equipment**

For latest updates visit www.rad.com

The IPVgate-20L SOHO analog VoIP gateway router is customer premises equipment (CPE) for the deployment of SIP-based VoIP services to SOHO and residential customers with ADSL2+ or Ethernet broadband connectivity.

Engineered for superior voice quality, the IPVgate-20L provides full telephone line replacement to allow fast and reliable access of SOHO and residential subscribers to next-generation VoIP services.

The product connects two FXS ports over any broadband Internet connection. The IPVgate-20L is fully interoperable with any standard SIP device.

One-box solution

RAD's SOHO analog VoIP gateway router is a one-box solution, providing cost-cutting SIP VoIP telephony service, broadband access, local area networking (LAN), and optional Wi-Fi access point via an IEEE 802.11b/g standard compliant wireless router.

Employing superior quality of service (QoS) technology and equipped with a powerful end-router, the gateway supports high quality communications for SOHO and residential analog telephones, fax machines, point-of-sale (POS) terminals, and PBXs to the VoIP service provider. Using the IPVgate-20L's dual LAN and Wi-Fi connectivity, Internet access can be shared in the SOHO environment by multiple terminals and wireless users.

Intelligent CPEs

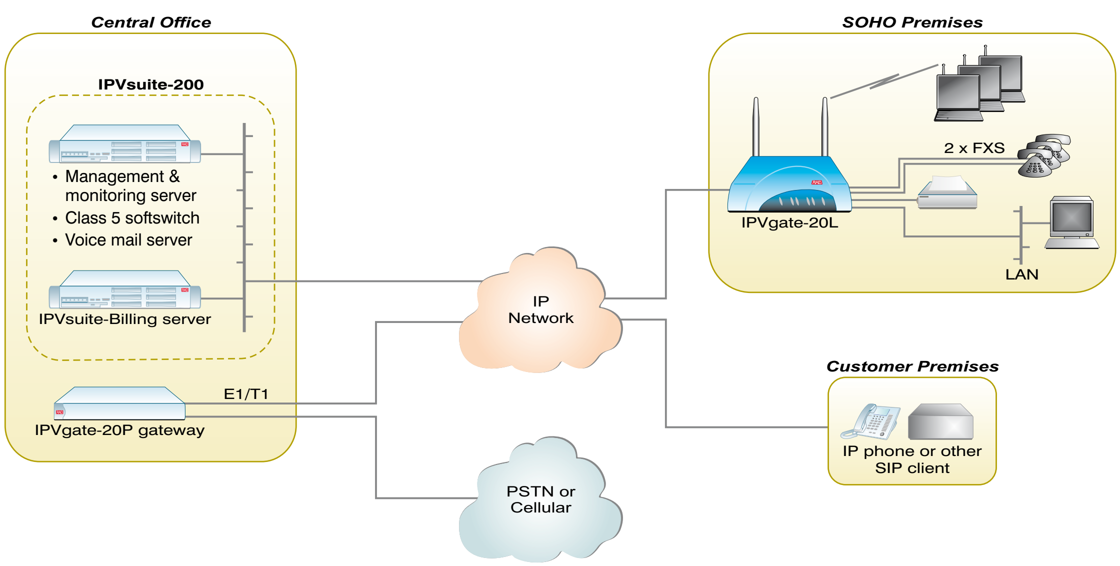
When operating as part of the RAD VoIP System (RVS), the IPVgate-20L works seamlessly with the IPVsuite-200 SIP softswitch and provisioning system, providing network statistics and analysis.

The intelligent IPVgate-20L gateway router is equipped with monitoring tools, enabling the central office to remotely configure, provision and manage units in service, thereby keeping Opex to a minimum.

Availability of service

In case of SIP server failure, the IPVgate-20L establishes peer-to-peer (P2P) phone calls to ensure high availability of service (AoS).

The IPVgate-20L is available with an integral ADSL modem, which provides ADSL line monitoring as well as automatic rerouting of all data via a backup router in the event of connectivity failure.



IPVgate-30

SMB IP-PBX and VoIP Gateway Router

The IPVgate-30 SMB IP-PBX and VoIP gateway router is customer premises equipment (CPE) for deployment of SIP-based VoIP services and enterprise PBX functionality to small and medium businesses (SMBs) with ADSL2+ or Ethernet broadband connectivity.

A robust end-router interoperable with any standard SIP device, including SIP IP phones and SIP clients, the IPVgate-30 serves as a PBX substitute to allow fast and reliable access to next-generation VoIP telephony combined with advanced PBX services.

The IPVgate-30 connects SIP-based IP phones, legacy analog telephones and faxes, POS terminals, SIP clients, and data over broadband access networks to SIP-based IP telephony networks. The device is fully interoperable with standards compliant SIP-based equipment.

Engineered for superior voice quality, the IPVgate product series has multilayered quality of service (QoS) mechanisms to ensure integrity of VoIP telephone calls. The ADSL line quality monitoring system in the intelligent IPVgate-30 CPE transmits information to the management and provisioning system in the IPVsuite-200 SIP softswitch for analysis.

Enterprise communications center

The IPVgate-30 creates an intelligent communications center, providing a one-box solution for enterprise communications. Supporting SIP IP phones and soft phones, the IPVgate-30 is both a VoIP gateway and router, providing broadband access and secure enterprise data networking by LAN and WLAN.

The product includes a built-in ADSL modem, router and firewall, enabling easy and secure access to the IP network. Analog fax machines connected to the IP-PBX also benefit from high quality T.38 support, ensuring reliable fax transmissions.

Full IP-PBX functionality

The robust integrated IP-PBX enables administrators to efficiently provision enterprise extensions and subscribers groups with easy to use and powerful rule management tools based on time, group, extensions, prefix, and services criteria. Complementing the IP-PBX functionality are tools for sound management, music on hold (MOH), Interactive Voice Recognition (IVR), personalized voice mail, queuing management, trunk management, and Call Details Record (CDR) generation, in addition to other features.

Intelligent CPEs

When operating as part of the RAD VoIP System (RVS), the IPVgate-30 works seamlessly with the IPVsuite-200's built-in management system, providing network statistics and analysis.

The intelligent IPVgate-30 is equipped with monitoring tools, enabling the central office to remotely configure, provision and manage units in service, thereby keeping OpEx to a minimum.

Interfaces

In addition to supporting SIP phones and soft phones over its WAN interface or the integrated four-port Ethernet switch, IPVgate-30 also connects two legacy analog telephones, faxes or POS terminals together with data over broadband to SIP-based IP telephony networks. Secure wireless networking is an optional feature.

- **Quick and easy installation**
- **Connects two FXS ports on an ADSL2+ (Annex A/B) or Ethernet connection**
- **Full-featured IP-PBX, including extension and group management, MOH, IVR, and personalized voice mail**
- **Built-in powerful router for single-box customer premises solution**
- **Four Ethernet ports with built-in switch for data connectivity in the LAN**
- **WLAN by Wi-Fi access (IEEE 802.11b and IEEE 802.11g)**
- **Supports advanced Class 5 call services, including caller ID, call on-hold, call waiting and transfer**
- **Remote management via IPVsuite-200 SIP softswitch ensures low total cost of ownership (TCO)**
- **Multilayered QoS for superior voice quality**

For latest updates visit www.rad.com

Availability of service

In case of SIP server failure, the IPVgate VoIP gateways establish peer-to-peer phone calls to ensure high availability of service (AoS).

The IPVgate-30 is available with an integrated ADSL2+ modem, which provides ADSL line monitoring as well as automatic rerouting of all data via a backup router in the event of connectivity failure.